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(54) Digital audio interface in a GSM device.

(57) A GSM mobile telephone comprises a digital audio interface for interfacing the system simulator (SS) for testing purposes. In the arrangement according to the invention a digital signal processor (11) and a codec (12) are implemented in the same

physical unit, whereby the system simulator (SS) interfaces to the telephone via a DAI block (14) and a JTAG data codec register (13) arranged in the DSP unit.

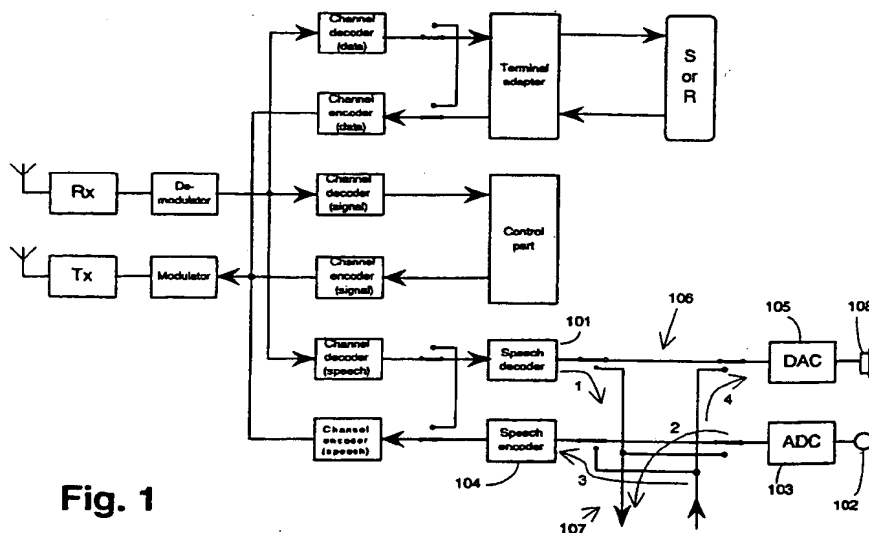


Fig. 1

The invention relates to a digital audio interface (DAI) in a GSM-device, according to the introduction of Claim 1. The digital audio interface in a GSM system (Global Mobile System) is a specified interface between a mobile station (MS) and a GSM system simulator (SS).

This interface is used in the conformance testing of a GSM device, especially in performing the bit exact test of the speech coder/decoder and in testing the sending/receiving loudness ratio (SLR/RLR). Fig. 1 represents the MS test arrangement of a mobile station in accordance with the GSM specification (GSM 11.10, Part III, Aspect III, Entry III.1: Special Conformance Testing Functions; Figure III.1-1/GSM 11.10: Test loops in the MS), wherein the essential parts with respect to the invention are marked: in the receive direction speech coder 101, digital interface 106, digital-analog converter 105 and speaker 108, and in the transmit direction microphone 102, analog-digital converter 103, as well as digital interface 107 and speech coder 104. According to the specification the interfaces transfer 13 bits wide linear PCM data in serial mode at 104 kbits/s. It is also possible to interface, via digital audio interface 106/107, to the system simulator SS (not shown) to connect the speech data in both the transmit and receive directions. By means of the digital audio interface it is possible to observe and control the data stream in between the D/A and A/D conversion and the coding/decoding of the digital speech signal.

In Fig. 1 the internal test loops, i.e. test paths, of the MS device required by the specification are represented by reference numbers 1 to 4. Path 1 transmits decoded speech from speech decoder 101 of the MS device to simulator SS; this path is used for down link speech transcoding testing. Path 2 transmits a signal converted by A/D converter 103 and originating from acoustic input 102 to the simulator; this path is used for the acoustic devices test. Path 3 transmits a signal from simulator SS to speech encoder 104 of the MS device; this is used for up link speech transcoding testing. Finally, path 4 transmits a signal from the simulator to A/D converter 105 and to the acoustic speech output device (speaker); this path is used for the acoustic devices test.

In digital interface 106/107 (DAI), according to the specification represented in Fig. 1, the linking of the system simulator is a very clear function because the specification is based on the assumption that the D/A and A/D converters on one hand, and the coders/decoders on the other hand, are separate physical devices with interface DAI in between.

However, the applicant has, for different reasons, reached a more cost-effective solution in realizing the mobile station MS, wherein D/A and

A/D converters and coders/decoders are implemented in one physical unit. This type of unit comprises a DSP processor for the speech transcoding function and an analog codec for the D/A and A/D conversion of the speech signal, as well as codec data register CDX, through which the system simulator SS has to interface to the MS device in order to meet the DAI specifications.

However, this creates problems. One alternative is that the DSP processor reads and writes into the data register CDX and subsequently interfaces to the system simulator via a serial SIO interface (not shown). However, this means that a specific program would have to be installed in the DSP processor that takes care of transferring the speech data from the SIO port to the codec data register and vice versa. This would result in a disadvantageous situation, where the MS would not be working in normal mode (i.e. would not be working with the normal mode speech coding program) during testing. Furthermore, the use of the serial port SIO for testing purposes limits the potential future use of the port for other purposes.

This problem is solved according to the invention in such a way that a JTAG (Joint Test Action Group, IEEE 1149.1) test access port arranged in the DSP unit is used in a special way according to the characterizing portion of Claim 1.

The invention is described in the following with the aid of an exemplary embodiment with reference to the appended drawings, where:

Fig. 1 represents a test loop arrangement according to the GSM specifications, whereby the members interfacing to the DAI interface and the required test paths are marked with reference numbers;

Fig. 2 represents a block diagram of the DAI arrangement according to the invention with the implementation of the test paths according to Fig. 1; and

Fig. 3 is a more detailed block diagram of the DAI block.

The basis for this above-described embodiment is a mobile device, i.e. a digital radio telephone MS for which a digital signal processor unit DSP16C was selected arranged on one chip and specified by the applicant. According to Fig. 2 this unit comprises DSP processor 11 for the speech transcoding function, analog codec (CODEC) 12 for the D/A and A/D conversion of the speech signal, and codec data register (CDX) 11a as well as JTAG codec data register (JCDX) 13. Test data input TDI and test data output TDO are interfaced to the JCDX and they interface the MS to the system simulator SS via DAI block 14.

Data from DSP processor 11 can be transmitted to JCDX register 13 via codec data register 11a, and, on the other hand, from codec 12. Cor-

respondingly, register 13 can write in either direction, that is, to register 11a and codec 12. Fig. 2 also shows how test paths 1 to 4 of the specification according to Fig. 1 are implemented in the DAI arrangement according to the invention.

The set of commands of the DSP processor further includes two special JTAG commands which make it possible to determine from which direction JCDX register 13 reads and where it writes. The "Program Test" command directs the JCDX register to read path 1 and to write path 3 towards processor 11; thus codec 12 is not connected to the JCDX register.

Correspondingly, the "Codec Test" command directs the JCDX register to read path 2 and write path 4 towards codec 12; thus processor 11 is not connected to the JCDX register.

The above-mentioned commands can be given from the system simulator SS via DAI block 14.

The essential feature of the invention is that the DAI interface is completely hardware-based. The interface is marked in Fig. 2 by line B - B. Consequently no special test configuration is necessary in the radio telephone to implement paths 1 to 4. No test program for testing needs to be loaded in the processor either. Only the two commands mentioned are needed for the DAI arrangement. When a command related to JCDX register 13 is given via the DAI block the hardware configuration of the DSPC16C unit is modified in order to form the test paths shown in Fig. 2.

Fig. 3 shows the construction of DAI block 14 in more detail. DAI interface B-B via which the MS interfaces to the SS via five leads is shown on the left in the figure. Correspondingly interface A-A (also in Fig. 2) via which the DAI block interfaces to JTAG data codec register (JCDX) 13 of the DSP processor is on the right. DAI block 14 comprises a control block (CONTROL) which controls the transfer of data between the SS data registers on the left in the figure and the JTAG data registers on the right. The data transfer is controlled by clock lines "SS clock" and "TCK". On the top of Fig. 3 there are control lines "test cont 1" and "test cont 2" which come from the system simulator on the left and which enable the SS to control the DAI block and thus also JTAG register 13 to different states via block "Select instruction". In other respects the markings in the block diagram in Fig. 3 are clear to those skilled in the art.

Claims

1. A digital audio interface (DAI) which enables the interfacing, to the system simulator, of the output signal of the device speech decoder (101), the signal from the acoustic input (102) of the device after the analog-digital conver-

sion (103), and the speech signal from the system simulator to the speech coder (104) of the device and to the digital-analog converter (105), whereby the speech coder (104), the speech decoder (101), the digital-analog converter (105) and the analog-digital converter (103) as well as the interfaces in between (106, 107) are implemented in the form of a circuit unit of the digital signal processor (11) and the codec (12), characterized in that a codec data register (11a) which functions as the interface between the signal processor and the codec (12) has, in addition, been installed in the signal processor (11), and a JTAG codec data register (13) between the codec data register (11a) and the test data input (TDI) and the test data output (TDO), whereby the test data input and the test data output (TDI, TDO) interface to the system simulator (SS) via the digital audio interface block (14).

2. A digital audio interface according to Claim 1, characterized in that commands which specify the reading and writing directions (1, 2, 3, 4) of the data of the JTAG codec data register (JCDX) are added to the set of commands of the signal processor (11).
3. A digital audio interface according to Claim 2, characterized in that said commands can be given from the system simulator (SS) via the JTAG codec data register (13).

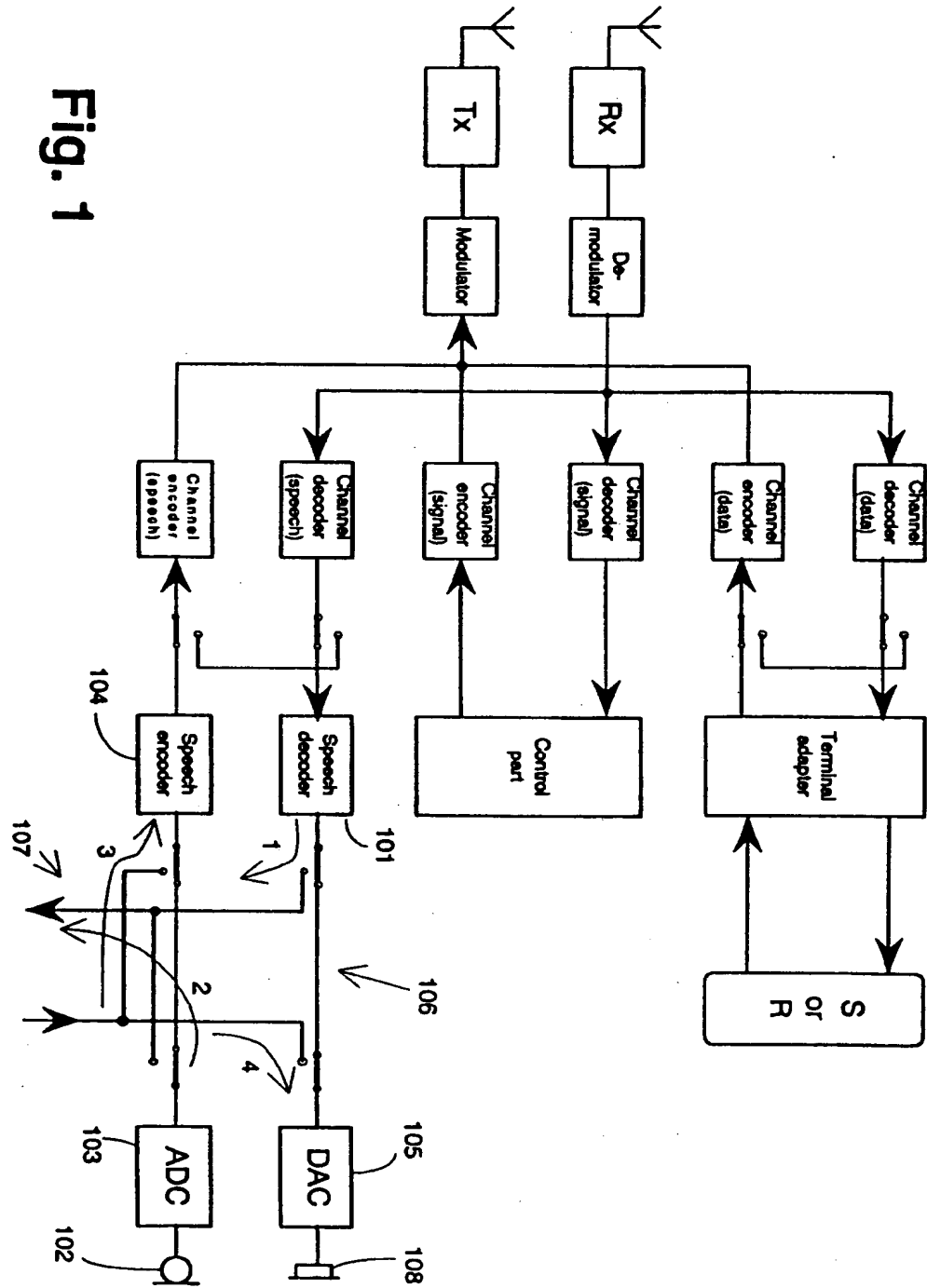


Fig. 1

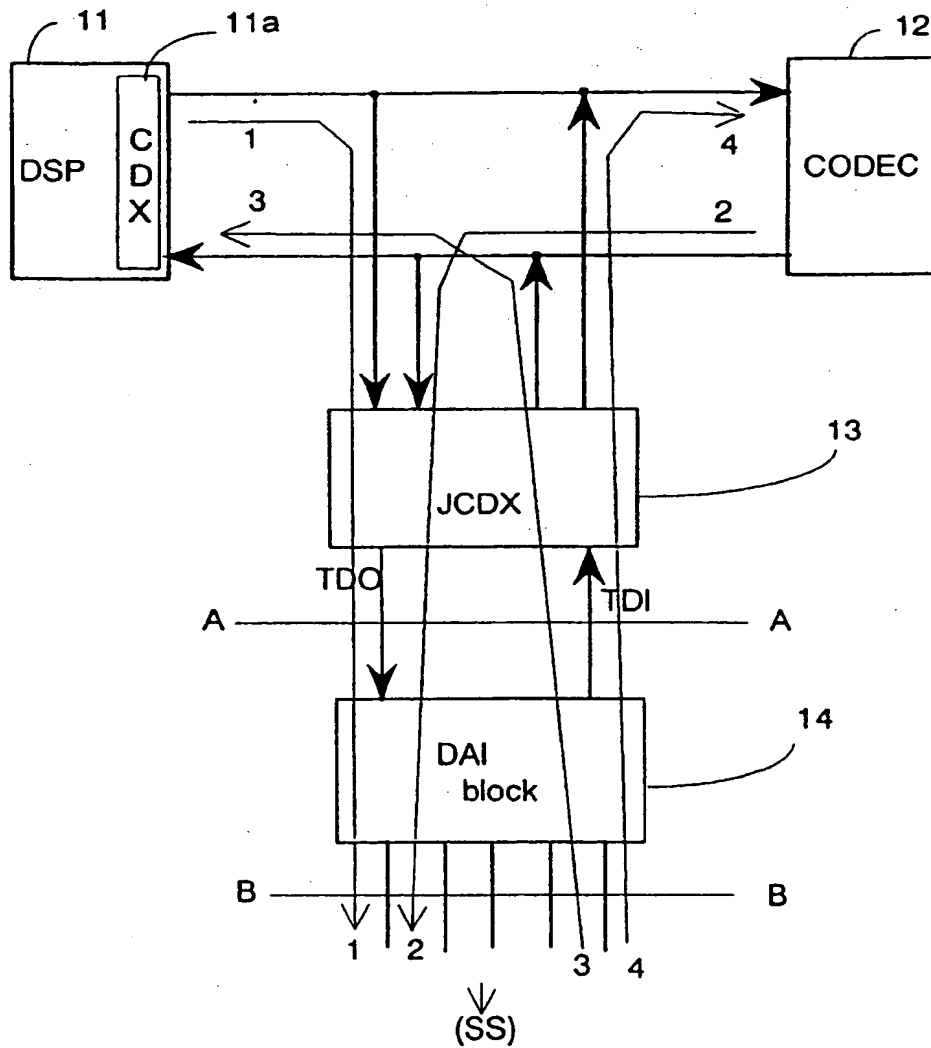


Fig. 2

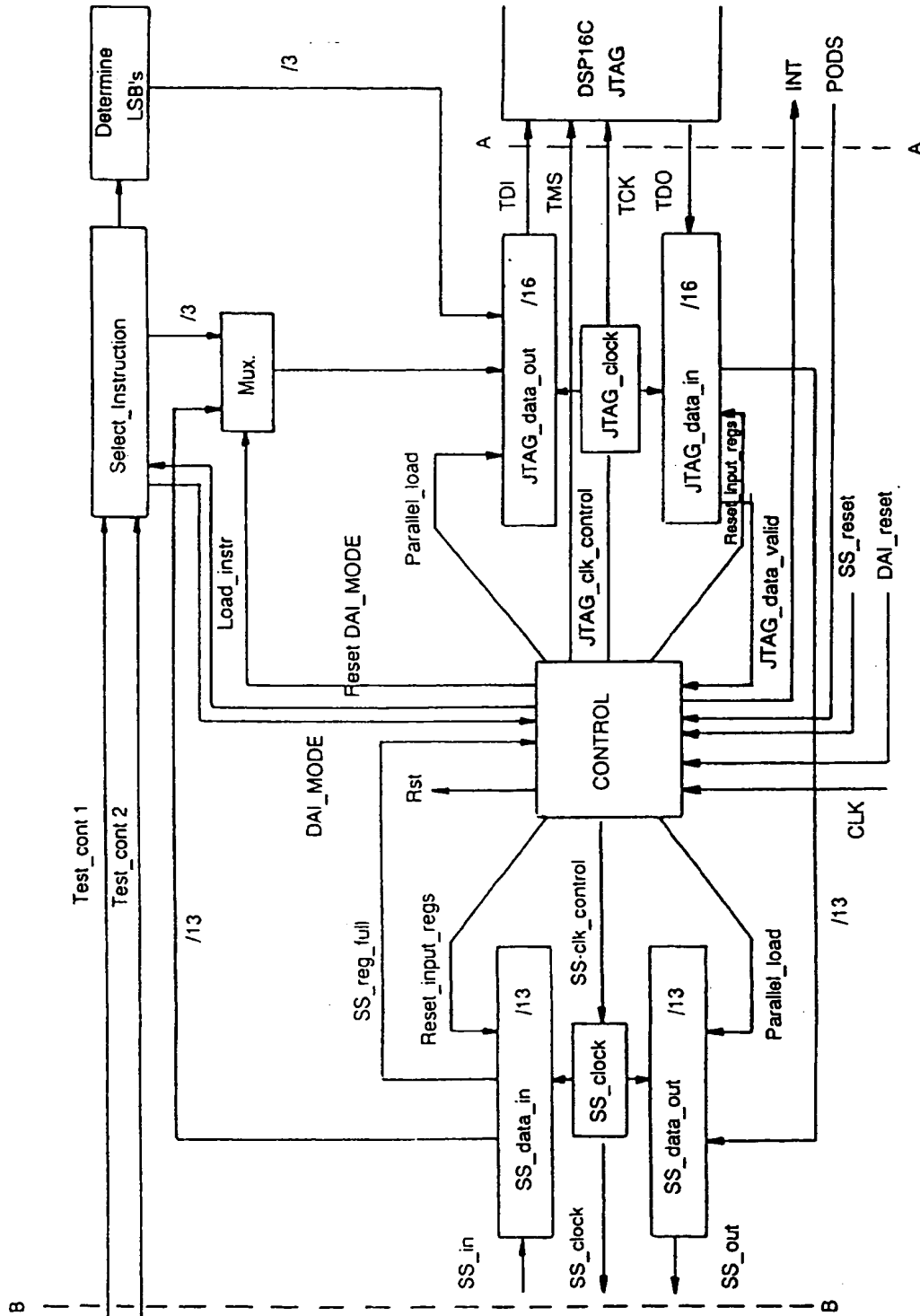


Fig. 3

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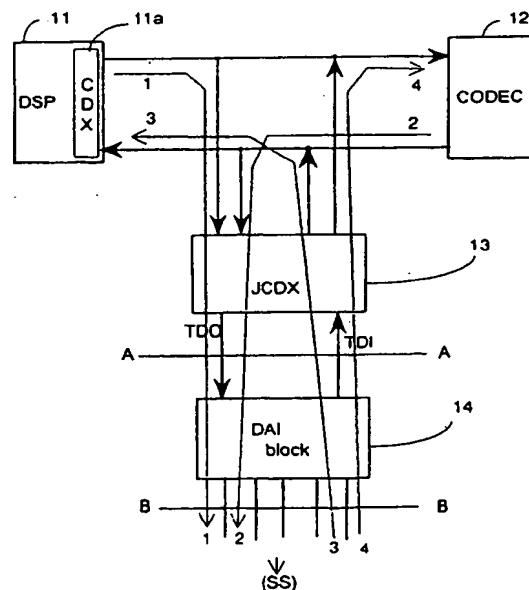
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07.07.93 Bulletin 93/27(71) Applicant: **NOKIA MOBILE PHONES LTD.**
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**Fig. 2**



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EUROPEAN SEARCH REPORT

Application Number

EP 92 30 8569

DOCUMENTS CONSIDERED TO BE RELEVANT

Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (Int. Cl.5)
A	US-A-5 023 902 (ANDERSON C.R.) * column 1, line 40 - line 46 * * column 4, line 11 - line 31 * ---	1-3	H04B17/00 H04Q7/04
A	PATENT ABSTRACTS OF JAPAN vol. 8, no. 173 (E-259)(1610) 9 August 1984 & JP-A-59 66 233 (NIPPON DENKI K.K.) 14 April 1984 * abstract * ---	1-3	
A	CONFERENCE PROCEEDINGS ON AREA COMMUNICATION, JUNE 13-17 1988 STOCKHOLM, SWEDEN, pages 478 - 481 JAN KALL 'THE GSM SYSTEM SIMULATOR' * The whole document * -----	1-3	
			TECHNICAL FIELDS SEARCHED (Int. Cl.5)
			H04B H04Q
The present search report has been drawn up for all claims			
Place of search THE HAGUE		Date of completion of the search 06 MAY 1993	Examiner CANALI F.
CATEGORY OF CITED DOCUMENTS			
X : particularly relevant if taken alone Y : particularly relevant if combined with another document of the same category A : technological background O : non-written disclosure P : intermediate document			
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